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Moving Image and Sound: Basic Issues and Training

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Pulse-code modulation: Center of the digital audio transition

I. A brief introduction to pulse-code modulation

Pulse-code modulation (PCM) is an encoding method to represent analog sound signals in a digital format. The term "pulse-code modulation" is used to refer to both the method of encoding and the product; that is, pulse-code modulation is both the method by which a digital signal is coded and the digital audio bitstream that results. PCM has several subtypes, including compressed and uncompressed variants. While digital transmission typically relies on compressed subtypes of PCM, it is in its uncompressed "linear" format (LPCM) that PCM becomes a preservation-level archival tool for storage and reformatting.[1] LPCM, a process that stores its digital signal with no loss of information, is referenced as an archival master standard codec by major archives and professional groups such as the National Archives,[2] the Library of Congress,[3] and the International Association of Sound and Audiovisual Archives;[4] it is so common in those contexts that the type and subtype (PCM and LPCM) have become essentially

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[1] Library of Congress, "Linear Pulse Code Modulated Audio (LPCM)," *Sustainability of Digital Formats,* 17 Feb. 2017, https://www.loc.gov/preservation/digital/formats/fdd/fdd000038.shtml, accessed 16 Oct. 2017.
[2] "Digitizing Motion Picture Film: Exploration of the Issues and Sample SOW," *Federal Agencies Digitization Guidelines Initiative*, The FADGI Audio-Visual Working Group, 18 Apr. 2016, http://www.digitizationguidelines.gov/guidelines/FilmScan_PWS-SOW_20160418.pdf, accessed 16 Oct. 2017.
[3] Library of Congress, "LPCM."
[4] International Association of Sound and Audiovisual Archives Technical Committee, "IASA-TC 04: Guidelines on the Production and Preservation of Digital Audio Objects," *IASA,* 2009, https://www.iasa-web.org/tc04/key-digital-principles, accessed 6 Dec. 2017, section 2.2.

interchangeable.[5] The Library of Congress describes LPCM as "comparable in transparency to uncompressed bit-mapped images."[6]

PCM today is typically encountered within an audio wrapper or container usually used with uncompressed audio, such as Waveform Audio File Format (WAVE)[7] or Audio Interchange File Format (AIFF);[8] it may also be encountered as the audio codec in a preservation-level multimedia container format such as Material Exchange Format (MXF)[9] or Matroska.[10] PCM is also the method of encoding audio for compact discs (CDs), Digital Audio Tape (DAT), and digital video discs (DVDs);[11] Blu-ray discs;[12] Digital Video (DV);[13] and the High-Definition Multimedia Interface (HDMI)[14] and the Audio Engineering Society 3 (AES3)[15] cable transmission standards.

II. History of pulse-code modulation

Historical timeline

While pulse-code modulation had been introduced in other fields, the first to incorporate the technique to an audiovisual field was Alec Reeves. Reeves, a British engineer, was the first

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[5] Library of Congress, "PCM, Pulse Code Modulated Audio," *Sustainability of Digital Formats,* 16 Feb. 2017, https://www.loc.gov/preservation/digital/formats/fdd/fdd000016.shtml, accessed 16 Oct. 2017.
[6] Library of Congress, "LPCM."
[7] Library of Congress, "WAVE Audio File Format," *Sustainability of Digital Formats,* 27 July 2017, https://www.loc.gov/preservation/digital/formats/fdd/fdd000001.shtml, accessed 16 Oct. 2017.
[8] Library of Congress, "Audio Interchange File Format (AIFF)," *Sustainability of Digital Formats,* 27 July 2017, https://www.loc.gov/preservation/digital/formats/fdd/fdd000342.shtml, accessed 16 Oct. 2017.
[9] Library of Congress, "Material Exchange Format (MXF)," *Sustainability of Digital Formats,* 28 Feb. 2017, https://www.loc.gov/preservation/digital/formats/fdd/fdd000013.shtml, accessed 16 Oct. 2017.
[10] Library of Congress, "Matroska Multimedia Container," *Sustainability of Digital Formats,* 28 Feb. 2017, https://www.loc.gov/preservation/digital/formats/fdd/fdd000011.shtml, accessed 16 Oct. 2017.
[11] Library of Congress, "PCM."
[12] "Why Blu-ray?," *Blu-ray Disc*. Blu-ray Disc Association, 2017, https://us.blu-raydisc.com/latest_news/why-blu-ray/, accessed 17 Oct. 2017.
[13] Roger Jennings, "How DV Works," Adaptec, Inc., 1997, https://web.archive.org/web/20071206032412/http://seaspray.trinity-bris.ac.uk/~altwfaq/graphics/video/1394/1394formats.html, accessed 6 Dec. 2017.
[14] "Knowledge Base," High Definition Multimedia Interface, 2017, https://www.hdmi.org/learningcenter/kb.aspx?c=11, accessed 17 Oct. 2017.
[15] European Broadcasting Union, "Specification of the Digital Audio Interface (The AES/EBU interface)," 2004, https://tech.ebu.ch/docs/tech/tech3250.pdf, accessed 6 Dec. 2017, 7.

to patent the technique of pulse-code modulation for communication by voice in 1939.[16] However, the first use of PCM to quantize and transmit speech was not until July 15, 1943, when the SIGSALY encryption system (developed by Bell Labs) transmitted an encrypted phone call between the Pentagon in Arlington, Virginia, and London.[17] The next major step in the application of PCM was in 1967, when the Technical Research Laboratory of Japan developed a PCM audio recorder for the NHK broadcast network.[18] Heitaro Nakajima, who was head of the team, recalled that they combined the PCM's "long-distance digital transmission" technology with computer storage techniques, a videotape recorder, and a digital-to-audio convertor; the initial result was a digital tape recorder that was too bulky and expensive to use, but that recorded high-quality sound.[19] Their use of a helical videotape recorder as storage, however, would persist for the next thirty years.[20]

After initial uses in telephony and forays into storage mechanisms, PCM was incorporated into television broadcasting. In 1972, in an attempt to improve their transmission quality, the British Broadcasting Corporation began sending digital transmissions digitally from their broadcast center to regional transmitters, who then decoded the audio for analog transmission to homes.[21] PCM also became the norm in telephony, where quality could be compressed or low without much consequence, when the American telephone system introduced digital switching (that is, connecting telephone lines) using PCM in 1976.[22] PCM audio on

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[16] Library of Congress, "PCM."
[17] "SIGSALY," Audio Engineering Society, 31 Jan. 2006, http://www.aes.org/aeshc/docs/recording.technology.history/sigsaly.html, accessed 17 Oct. 2017.
[18] Thomas Fine, "The Dawn of Commercial Digital Recording," *ARSC Journal* vol. 39, no. 1 (Spring 2008), 2.
[19] Heitaro Nakajima, oral history with William Aspray, Engineering and Technology History Wiki, http://ethw.org/Oral-History:Heitaro_Nakajima, accessed 6 Dec. 2017.
[20] Fine, "The Dawn of Commercial Digital Recording," 2.
[21] Fine, "The Dawn of Commercial Digital Recording," 2.
[22] Bernhard E. Keiser and Eugene Strange, eds., *Digital Telephony and Network Integration* (New York: Springer Science and Business Media, 1985), 4.

commercial videotape stock became widespread by 1978,[23] though widespread commercial adoption of PCM in recording (after NHK's initial recorder) only came in 1982 with the CD, which famously stores 80 minutes of stereo audio at a 44.1 kilohertz sampling frequency and 16-bit resolution.[24] CDs and most of the media that used PCM still exist today; DAT,[25] DV,[26] DVDs,[27] Blu-ray discs,[28] HDMI,[29] and AES3[30] standards all currently specify the ability to use pulse-code modulation.

Technological capabilities and early user groups

One of the first academic papers on PCM appeared in 1947 in the Institute of Electrical and Electronics Engineers, describing its novel (now basic) sampling and quantizing principles. The authors noted another novel capability: the division of channel transmissions by time. This feature meant that multiple PCM channels could transmit over the same frequency band, pulse by pulse, and take advantage of all previously-empty time available on the band.[31]

One of the initial selling points of PCM was its ability to transmit without interference. A paper published in 1948 in the Proceedings of the Institute of Radio Engineers touted PCM as a transmission system that could substantially improve the signal-to-noise ratio with its "rugged" defense against interference—that is, its ability to perform time-dimension multiplexing.[32]

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23 Sam Brylawski, Maya Lerman, Robin Pike, Kathlin Smith (eds.), *ARSC Guide to Audio Preservation* (Association for Recorded Sound Collections/Council on Library and Information Resources/Library of Congress, May 2015), 7.
24 Fine, "The Dawn of Commercial Digital Recording," 3, 13.
25 Library of Congress, "PCM."
26 Jennings, "How DV Works."
27 "DVD Studio Pro 4 User Manual," *Apple Documentation*, Apple Inc., Nov. 2009, https://documentation.apple.com/en/dvdstudiopro/usermanual/index.html#chapter=5%26section=3%26tasks=true, accessed 17 Oct., 2017.
28 "Why Blu-Ray?," Blu-ray Disc Association.
29 "Knowledge Base," HDMI.
30 European Broadcasting Union, "Specification of the Digital Audio Interface," 2004, 7.
31 H.S. Black and J.O. Edson, "Pulse Code Modulation," *Institute of Electrical and Electronics Engineers* vol. 60 (1947), 895-899: 896..
32 B.M. Oliver, J.R. Pierce, and C.E. Shannon, "The Philosophy of PCM," *Proceedings of the IRE* vol. 36, no. 11 (Nov. 1948), 1324-1331: 1328-9.

Though FM (or Frequency Modulation, its competing transmission technique at the time) was also resistant against interference, its signal varied only with frequency and left many small time gaps in its frequency band that could not be filled by pulses.[33] PCM could do this because it was a digital format, not analog, which the paper explains to its audience as follows:

> "In analogue machines the numbers involved are represented as proportional to some physical quantity capable of continuous variation…An increase in precision requires, in general, a proportional increase in the range of physical variables used to represent the numbers. Furthermore, small errors tend to accumulate and cannot be eliminated. In digital machines…precision increases exponentially with the number of digits, and hence with the size of the machine. Small errors, which are not large enough to carry any part from one state to another state, have no effect and do not cumulate."[34]

PCM was not just a new format or method, but a technology on the vanguard of one of the 20th century's most consequential shifts: the switch from analog to digital. It represented the new precision available with coded audio, and—as its use in SIGSALY and telephony infrastructure imply—its early user groups were audio engineers who needed a better way to communicate in the military, intelligence, telephony, and telemetry fields.

PCM also had implications for marketing commercial recordings. Most other common digital audio formats store audio in compressed form, such as the lossy Advanced Audio Coding (AAC)[35] and MPEG Layer III (MP3) formats, or the lossless Free Lossless Audio Codec (FLAC).[36] For this reason, many marketers treat uncompressed PCM as a selling point, touting the high quality in advertising copy; for example, a webpage for Blu-ray notes that it "requires that there is support for…linear PCM," and adds that the quality audio experience will mean "the hours spent ensuring that the soundtrack is seamless synced with the on screen action will not

[33] Oliver et al, "The Philosophy of PCM," 1329.
[34] Oliver et al, "The Philosophy of PCM," 1329-1331.
[35] Library of Congress, "Advanced Audio Coding (MPEG-2)" *Sustainability of Digital Formats,* 17 Feb. 2017, https://www.loc.gov/preservation/digital/formats/fdd/fdd000036.shtml, accessed 6 Dec. 2017.
[36] Brylawski et al, *ARSC Guide,* 11.

have been in vain for the sound technician."[37] Pulse-code modulation is currently the predominant method used to reformat analog audio in uncompressed digital form.

III. The process of pulse-code modulation

The fundamental difference between an analog and a digital audio signal is the range of values available to describe the signal. An audio signal is a representation of sound waves as frequency, which humans interpret as pitch, continuously sustained over time. Analog signals can take any of an infinite range of frequencies, including those well outside the range of human hearing. Digital signals, on the other hand, must set a range of measurable frequencies that does not correspond to the full range of the natural world, and must associate the frequencies they measure with discrete timestamps and amplitude levels.[38] Digital signals thus inherently lose information available in the analog world, though they can approximate analog quality by taking high numbers of samples measured in as wide a frequency range as possible. (It is worth noting that analog *carriers* have lower dynamic ranges than digital carriers, though both of them are lower than that of human hearing.[39]) The limiting factor in the analog-to-digital conversion is storage space, as digital audio encoded with high fidelity takes up enormous amounts of digital room.

The transformation between analog and digital signal via pulse-code modulation comprises two steps: sampling and quantization. The sampling rate and bit depth values associated with these processes are key to assessing the ensuing digital signal's fidelity. Sampling is the process of breaking the continuous time of the analog audio into discrete points.

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[37] "Why Blu-Ray?," Blu-ray Disc Association.
[38] Brylawski et al, *ARSC Guide,* 10-11.
[39] Brylawski et al, *ARSC Guide,* 11.

Digital audio cannot hold information about a sound wave for every single moment in time; instead, it must sample the value of a sound wave at distinct moments, determined by a sampling rate. Thus, the sampling rate (typically measured in number of samples per second) is a measurement of how often the digital signal measures the amplitude of the analog waveform.[40] By way of illustration, a sampling rate of 10 samples/second would indicate that the PCM process encoded ten samples of the analog waveform every second, thus losing all of the intermediate amplitude values between those timestamps.

Quantization is the process of converting the amplitude information gleaned during sampling into a series of stepped, discrete amplitude values. In the same way that digital audio cannot hold the full, continuous range of time that analog audio can, digital audio cannot hold the full, infinite range of possible frequencies available to analog audio. Instead, it must translate the actual audio frequency into one of a large number of possible amplitude levels. The number of levels is determined by the bit depth, and a larger bit depth indicates more levels. (The exact number of levels is equal to 2^x, where *x* is the bit depth; a waveform sampled at 8 bits per sample would have 2^8 levels, or 256–128 positive and 128 negative).[41]

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[40] Keiser and Strange, *Digital Telephony,* 25.
[41] Keiser and Strange, *Digital Telephony,* 25.

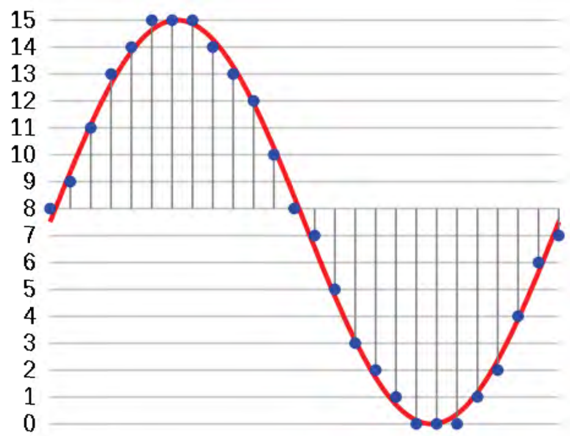

Fig. 1. The analog audio (red line) is encoded and stored as digital samples (blue dots), discrete in both time (x-axis) and amplitude (y-axis).[42]

Linear PCM (LPCM) is what results from this process once the process of sampling and quantization are done; the digital LPCM signal records the unaltered time and amplitude values of each sample, without any conversion or prediction. As humans cannot hear digital audio, any playback must involve digital-to-audio conversion. This process involves identifying the beginning and end bits in each sample, as well as those samples' quantization levels, after which the same encoding algorithm is applied in reverse.[43]

III. Drawbacks, potential for error, and competitors

PCM is typically encoded in its linear form as uncompressed digital audio. An analog audio signal encoded using PCM at a very low sampling rate and a very low bit depth may result in a digital signal of unsatisfactory quality when compared to the original analog. Several

[42] Wikimedia Commons contributors, "File:Pcm.svg," Wikimedia Commons, the free media repository, https://commons.wikimedia.org/w/index.php?title=File:Pcm.svg&oldid=264826365, accessed December 6, 2017

[43] Keyur Desai, "PCM Encoding and Decoding" (lecture slides, Michigan State University Department of Electrical and Computer Engineering, East Lansing, MI, 21 Mar. 2007), http://www.egr.msu.edu/classes/ece458/radha/ss07Keyur/Lab-Handouts/Lab9PCM.pdf, accessed 6 Dec. 2017.

variations on PCM do allow the compression of audio and are used primarily in telephony, where bandwidth is crucial. μ-law and A-law PCM use compressed logarithmic quantization, rather than uncompressed linear; the former is used in North American telephony, while the latter is European.[44] Forms of PCM that use predictive encoding strategies can also reduce bandwidth. Differential PCM (DPCM) stores sample values by measuring the difference between adjacent samples' quantization levels.[45] Adaptive DPCM (ADPCM) is a similar strategy that, prior to storing the difference between two samples' quantization levels, also adjusts the scale of the difference by adjusting the size of the actual step between all quantization levels. ADPCM bases this adjustment on previous variance.[46] Both variations on PCM allow lower bit depths while introducing further uncertainty; storing the difference between two samples in lieu of storing the actual sample amplitude requires a lower bit depth and thus less storage space, but it is a predictive technique that may drift from the absolute amplitude values.

PCM does presents the possibility of error in the encoding process. Quantization error is inherent, as an amplitude value must be rounded to conform to the nearest discrete quantization level.[47] Sampling error is not inherent, but quite possible to provoke; the association of samples with specific timestamps means that encoding and decoding the audio is dependent on extremely accurate timing.[48] The selection of appropriate sampling rates also plays a role; while it is not necessary to conform to the highest sampling rates as listed above, undersampling may provoke aliasing, or the appearance that a wave has a frequency lower than its actual frequency. This misinterpretation loses information about the original wave and may introduce artifacts.[49] Those

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[44] Keiser and Strange, *Digital Telephony,* 27.
[45] Keiser and Strange, *Digital Telephony,* 41.
[46] Keiser and Strange, *Digital Telephony,* 47.
[47] Brylawski et al, *ARSC Guide,* 11.
[48] Desai, "PCM Encoding and Decoding."
[49] Bruno A. Olshausen, "Aliasing" (handout, PSC 129: Sensory Processes, University of California at Berkeley, Berkeley, CA, 10 Oct. 2000), http://redwood.berkeley.edu/bruno/npb261/aliasing.pdf, accessed 6 Dec. 2017.

digitizing audio must adhere to the Nyquist-Shannon Sampling Theorem, which states that the sampling rate must be twice the highest recorded frequency in the original wave. A sampling rate this high will be sure to capture the outline a full wave for even the highest-frequency (shortest) wave on the x-axis. As the human ear can hear up to about 20 kHz, sampling rates should typically be at least 40 kHz.[50]

Though PCM has emerged as the widely-accepted standard, it is not the only method of digital encoding and decoding of audio. One alternative is pulse-density modulation (PDM), which samples audio using only one bit but a very high sampling rate. This method is simpler to carry out but results in less refined digital audio; PCM's advantage for those who are able to increase complexity is that it is straightforward and easy to controls.[51] Other, more well-known, competitors of PCM are the common lossy and losslessly compressed digital encoding formats, including AAC, MP3, and FLAC. MP3 and AAC are both widely adopted (the former with casual users, the latter on streaming sites and mobile devices), and both have the distinct advantage of producing much smaller yet relatively high-quality files files. FLAC is perhaps the only codec that can compete with PCM in terms of quality; it is stored as a compressed file but can be decoded to the full-quality original with no loss of information. FLAC is used as an alternative in institutions that want a lossless file but want to save storage space.[52] However, engineers who want a full-quality file will typically turn to PCM because of its familiarity and its simple, uncompressed structure.

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[50] Brylawski et al, *ARSC Guide,* 10.
[51] Thomas Kite, "Understanding PDM Digital Audio." Beaverton, OR: Audio Precision, Inc., 2012, 3.
[52] "Introduction," Free Lossless Audio Codec, Xiph.Org Foundation, 2014, https://xiph.org/flac/features.html, accessed 6 Dec. 2017.

IV. Current archival uses of pulse-code modulation

The early users of PCM were largely engineers. Until CDs became widespread in the 1980s, pulse-code modulation was thought of as almost exclusively a communications mechanism. In fact, one 1985 textbook reminds the reader: "While PCM is designed primarily to handle voice, occasions arise in which a subscriber may use a voice line for data transmission via modems,"[53] not imagining that the reader could need it for another purpose. However, another major user group for PCM is the cultural heritage sector, which includes anyone who wants to digitize audio recordings to professional standards.

The key factor in making LPCM audio an excellent candidate for cultural heritage standards, as well as for commercial success, is its fidelity. For the consumer, this translates to enjoyment of good listening quality; for the archivist, this translates to the closest quantifiable reproduction of the original analog signal. LPCM records the unaltered time and amplitude values for each sample, unlike DPCM or ADPCM (both of which only store difference, not absolute value, of amplitude, and the latter also changes the scale in the process). Because the samples are linear in terms of timestamps, and because the distance between quantization levels is always the same, LPCM audio is relatively simple to decode by trial and error, even if encountered out of context or out of a typical wrapper.[54] The Library of Congress compares it to an "uncompressed bit-mapped image" in terms of simplicity,[55] while a European-United States joint working group on audio collections had this to say in a report:

> "Linear PCM data, irrespective of sampling rate, word length, method of packing data into bytes and left-to-right or right-to-left arrangement of bits and bytes, can be decoded

[53] Keiser and Strange, "Digital Telephony," 31.
[54] DELOS/NSF Working Group, "Working Group on Spoken-Word Audio Collections," European Research Consortium for Informatics and Mathematics., May 2003, https://www.ercim.eu/publication/ws-proceedings/Delos-NSF/SpokenWord.pdf, accessed 16 Oct. 2017, 34.
[55] Library of Congress, "LPCM."

> by relatively simple trial-and-error, and we can expect this to be the case indefinitely. PCM is in this sense a 'natural' representation for audio, and has very good long-term prospects regardless of the remaining problems of format migration."[56]

The format is so widely accepted that a European Union-funded project exploring the feasibility of standardizing audiovisual files is considering three different video codecs for the final profile, but only uncompressed LPCM as the audio codec. In a listserv message justifying the decision, project member Emanuel Lorrain wrote that LPCM was chosen "because it is an established and well adopted standard in the AV archiving community (ie. TC-04) and because it is very practical, simple, interoperable [*sic*], etc."[57]

The cultural heritage sector has also adopted a wrapper that manages to retain this uncompressed quality while adding context in the form of metadata. Though LPCM is simple, it does not actually include any metadata about its encoding in the stream itself, and a PCM stream out of context would be difficult (though not impossible) to decode. Therefore, values such as the sample rate, bit depth, number of channels, and other technical metadata are stored in a standard wrapper. In archival settings, this wrapper is usually Broadcast WAVE, a format based on WAVE (Waveform Audio File Format) that can hold uncompressed audio. Broadcast WAVE, or BWF, was developed by the European Broadcasting Union to include a header and "chunks" where descriptive, technical, and administrative metadata may be embedded.[58] The embedded metadata will remain associated with the file even if it moves between folders, computers, or systems. LPCM wrapped in BWF is the archival master format for audio at the Library of

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[56] DELOS/NSF Working Group, "Working Group on Spoken-Word Audio Collections," 34.
[57] Emanuel Lorrain, "Re: Announcing CELLAR (Codec Encoding for LossLess Archiving and Realtime transmission)," *AMIA-L* listserv, 2 Dec. 2015.
[58] Library of Congress, "Broadcast WAVE Audio File Format, Version 1," *Sustainability of Digital Formats,* 27 July 2017, https://www.loc.gov/preservation/digital/formats/fdd/fdd000356.shtml, accessed 6 Dec. 2017.

Congress,[59] and is recommended by IASA[60] as well as a joint American-European working group for archival purposes.[61]

V. Technical specifications and fidelity

Commercial PCM specifications for discs (e.g. DVD, CD) and HDMI uniformly support sampling rates of 48 to 96 kHz and bit depths of 16 to 24.[62] As PCM moves from storage on optical media to storage in digital files, analog-to-digital converters have expanded the possibilities for high-quality transfers, and sampling rates up to 192 kHz at 24-bit depth are now widely available. The audiovisual cultural heritage sector has traditionally had to provide support for de facto or de jure standards from the commercial world, as many collections include large numbers of born-digital consumer materials. However, these standards are often well below the range of possibility. In the case of analog materials, institutions are also motivated to capture as much detail of the original signal as possible, and these high sampling rates may be a new standard for the field. In a post to the Association of Moving Image Archivists listserv, archivist Reto Kromer described the difference that sampling at 192 kHz made in an attempt to recover a "highly decomposed" optical soundtrack on an early-1930s film and argued that the standard is appropriate for an archives wishing to do sound restoration.[63]

As of 2017, the Federal Agencies Digital Guidelines Initiative Audio-Visual Working Group, in its specifications for MXF as an audiovisual preservation format, notes that "many archiving organizations strongly endorse linear PCM encoding" with a 48kHz sampling rate and

[59] Library of Congress, "Broadcast WAVE."
[60] IASA, "IASA-TC 04," section 2.8.2.
[61] DELOS/NSF Working Group, "Working Group on Spoken-Word Audio Collections," 22.
[62] "Knowledge Base," HDMI; "DVD Studio Pro 4 User Manual." Apple Documentation; Brylawski et al, *ARSC Guide.*
[63] Reto Kromer, "Re: [AMIA-L] Audio and Video Digitization Specifications," *AMIA-L* listserv, 27 Apr. 2016.

bit depth of 24.[64] This archival standard, as articulated by the United States government, is higher than the audio CD standard of 44.1 kHz/16-bit set in the Philips and Sony *Red Book* of compact disc specifications.[65] However, while an International Association of Sound and Audiovisual Archives white paper on audio preservation standards recommends a minimum of 48kHz/24 bit, it notes that in "heritage/memory institutions" 96 kHz/24 bit "has become widely adopted."[66]

Though standards do not always align across cultural heritage institutions, the full gamut of sampling rate and bit depth values for commercial audiovisual formats is far wider. For example, while Digital Video (DV) is sampled as low as 32 kHz/12 bit,[67] sampling rates can reach 192 kHz and 24 bits for MXF.[68] However, many wrappers can accommodate multiple types of PCM and even multiple audio codecs. The WAVE file format and the Broadcast WAVE format it generated, for example, can wrap PCM at 8 and 16 bits as well as compressed MPEG formats.[69] These values are particularly interesting to compare with early uses of PCM; for example, the NHK network in Japan in 1967 produced audio with a 32 kHz sampling rate and 13-bit depth, not far off from the aforementioned 32 kHz/12 bit DV standard.[70] While the possibilities have changed exponentially, the early specifications were robust enough to hold up to some standards today.

It is important to note that these coding specifications are most relevant in the case of

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[64] The FADGI Audio-Visual Working Group, "AS-07: MXF Archive and Preservation Format Application Specification," Federal Agencies Digitization Guidelines Initiative, 8 Sept. 2017, http://www.digitizationguidelines.gov/guidelines/AS-07_20170908.pdf, accessed 6 Dec. 2017, 32.
[65] Brylawski et al, *ARSC Guide,* 11.
[66] International Association of Sound and Audiovisual Archives Technical Committee, "IASA-TC 03: The Safeguarding of the Audio Heritage: Ethics, Principles and Preservation Strategy, version 3," IASA, December 2005, https://www.iasa-web.org/sites/default/files/downloads/publications/TC03_English.pdf, accessed 6 Dec. 2017, 8.
[67] Jennings.
[68] FADGI Audio-Visual Working Group, "AS-07," 32.
[69] European Broadcasting Union, "Tech 3285 v2: Specification of the Broadcast Wave Format (BWF)" (Geneva: European Broadcasting Union, May 2011), https://tech.ebu.ch/docs/tech/tech3285.pdf, 7, 11.
[70] Fine, "The Dawn of Commercial Digital Recording," 2.

born-analog materials. There is a mismatch between some born-digital audio and the specifications to which cultural heritage institutions are expected to digitize, and archivists must be careful to understand the original specifications. As audiovisual archivist Dave Rice pointed out in a 2012 Association of Moving Image Archivists listserv thread, transferring a CD (44.1 kHz/16-bit) at 96 kHz/24 bit will create “new sample values at points in time that aren’t represented in the original audio and will degrade the quality of the audio (although slightly),” and re-encoding at a higher bit depth will no longer match the original artifact.[71] IASA notes in another technical paper that preservation storage copies should equal the sampling rate and “at least equal” the bit depth of the original; this slightly different language is probably because the last two bits would have no content and be ignored.[72] In these cases, bit depth and sampling rates have limits; in the case of analog originals, there is likely a perceptible limit, but the audiovisual archiving community is more likely to hit a storage limit.

VI. Conclusions

Since its audio transmission debut in 1943, pulse-code modulation has expanded beyond its dominance in the communications sector to become a global current standard for digital audio recording and encoding today. Its widespread acceptance among audio engineers and cultural heritage institutions ensures that the codec will survive across user groups. With a simple structure and a standard wrapper to marry uncompressed (or compressed) content with metadata, PCM has no known preservation issues nor competitors and is likely to remain useful, standardized, and readable for some time. Furthermore, researchers are developing the PCM method for application in areas other than audio, including researched as a method to losslessly

[71] Dave Rice, “Re: More about digital video recording than you really want to know.,” *AMIA-L* listserv, 12 Dec. 2012.
[72] IASA, “IASA-TC 04,” sections 2.2-2.3.

compress images for hospitals[73] and patented to predict neighboring pixels in a high efficiency video codec.[74] With wide adoption, inclusion in standards, and active development, it seems likely that pulse-code modulation is as stable as a digital format can be.

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[73] Rime Raj Singh Tomar and Kapil Jain, "Lossless Image Compression Using Differential Pulse Code Modulation and its Application," *Institute of Electrical and Electronics Engineers 2015 International Conference on Computational Intelligence and Communication Networks* (Dec. 2015), 397-400.
[74] Wen Gao, Minqiang Jiang, Ye He, Jin Song, and Haoping Yu, US Patent 9813733 B2 (2017), United States Patent and Trademark Office.

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Black, H.S., and J.O. Edson. "Pulse Code Modulation." *Institute of Electrical and Electronics Engineers* vol. 60 (1947), 895-899.

- An early paper detailing the basic principles of pulse-code modulation and what the creators found noteworthy; a way to look at the origins of PCM using a primary source.

Brylawski, Sam, Maya Lerman, Robin Pike, and Kathlin Smith (eds.). *ARSC Guide to Audio Preservation*. Association for Recorded Sound Collections/Council on Library and Information Resources/Library of Congress, May 2015.

- A highly readable introduction to audio preservation, including basic discussion of audio principles and technological history as well as best archival practices for everything from accessioning to reformatting.

DELOS/NSF Working Group, "Working Group on Spoken-Word Audio Collections," European Research Consortium for Informatics and Mathematics., May 2003, https://www.ercim.eu/publication/ws-proceedings/Delos-NSF/SpokenWord.pdf, accessed 16 Oct. 2017

- A call to action for audiovisual collections that presents an overview of joint priorities in the field for American and European collections managers, librarians, and archivists; served as a point of evidence for international acceptance of PCM.

Desai, Keyur. "PCM Encoding and Decoding." Lecture slides from the Michigan State University Department of Electrical and Computer Engineering, East Lansing, MI, 21 Mar. 2007. http://www.egr.msu.edu/classes/ece458/radha/ss07Keyur/Lab-Handouts/Lab9PCM.pdf, accessed 6 Dec. 2017.

- A technical explanation of encoding and decoding; useful for its specificity about the decoding process rather than as an overview.

"DVD Studio Pro 4 User Manual." *Apple Documentation.* Apple Inc., Nov. 2009. https://documentation.apple.com/en/dvdstudiopro/usermanual/index.html#chapter=5%26section=3%26tasks=true. Accessed 17 Oct., 2017.

- One of the resources that include pulse-code modulation in discussions of industry specifications; many standards are proprietary or closed, so even the mention of PCM was helpful evidence.

European Broadcasting Union, "Specification of the Digital Audio Interface (The AES/EBU interface)," 2004, https://tech.ebu.ch/docs/tech/tech3250.pdf, accessed 6 Dec. 2017.

- Open specifications accompanied by thorough documentation addressing PCM as it is used in cables; dense and more appropriate for specifics than as appropriate as a jumping-off point for AES3 in general.

European Broadcasting Union. "Tech 3285 v2: Specification of the Broadcast Wave Format (BWF)." Geneva: European Broadcasting Union, May 2011, https://tech.ebu.ch/docs/tech/tech3285.pdf.

- Open specifications accompanied by thorough documentation are always appreciated, dense and more appropriate for specifics than as appropriate as a jumping-off point for BWF.

The FADGI Audio-Visual Working Group. "AS-07: MXF Archive and Preservation Format Application Specification." *Federal Agencies Digitization Guidelines Initiative*, 8 Sept. 2017, http://www.digitizationguidelines.gov/guidelines/AS-07_20170908.pdf. Accessed 6 Dec. 2017.

- A thorough set of standards for archivists who wish to digitize to widely-accepted guidelines; useful as evidence of the priorities of the cultural heritage field.

The FADGI Audio-Visual Working Group. "Digitizing Motion Picture Film: Exploration of the Issues and Sample SOW." *Federal Agencies Digitization Guidelines Initiative*, 18 Apr. 2016. http://www.digitizationguidelines.gov/guidelines/FilmScan_PWS-SOW_20160418.pdf. Accessed 16 Oct. 2017.

- A look into the issues of standardizing best practices for rich, analog audiovisual works and the pressure to get it right the first time; a resource for those who wish to see thought processes at work, and an interesting comparison of video to audio codecs (the audio standards are barely a question, while video codecs change from sample to sample).

Fine, Thomas. "The Dawn of Commercial Digital Recording." *ARSC Journal,* vol. 39, no. 1. Spring 2008, pg. 1-17.

- A succinct overview of early digital recording methods. His focus on recording is notable, as many resources on PCM were written in the era when it was largely used in communications and telephony.

Kite, Thomas. "Understanding PDM Digital Audio." Beaverton, OR: Audio Precision, Inc., 2012. 3.

- A succinct primer on a little-used technique; useful for those who want to encode audio off the beaten path. Hard to find, which of itself implied PDM has not found widespread adoption

Gao, Wen, Minqiang Jiang, Ye He, Jin Song, and Haoping Yu, US Patent 9813733 B2 (2017), United States Patent and Trademark Office.

- An example of further research and work being done with the PCM model in arenas other than audio; useful as an example of current work and prospective longevity of the encoding method.

International Association of Sound and Audiovisual Archives Technical Committee. "IASA-TC 03: The Safeguarding of the Audio Heritage: Ethics, Principles and Preservation Strategy, version 3." IASA, December 2005. https://www.iasa-web.org/sites/default/files/downloads/publications/TC03_English.pdf. Accessed 6 Dec. 2017.

- A thorough set of standards for archivists who wish to digitize to widely-accepted guidelines; helped me understand the priorities of the cultural heritage field as they relate to PCM.

International Association of Sound and Audiovisual Archives. "IASA-TC 04: Guidelines on the

Production and Preservation of Digital Audio Objects." IASA, 2009. https://www.iasa-web.org/tc04/key-digital-principles. Accessed 6 Dec. 2017.

- A thorough set of standards for archivists who wish to digitize to widely-accepted guidelines; helped me understand the priorities of the cultural heritage field as they relate to PCM, and the role that access to and storage of uncompressed audio plays in cultural preservation.

"Introduction," Free Lossless Audio Codec, Xiph.Org Foundation, 2014, https://xiph.org/flac/features.html, accessed 6 Dec. 2017."

- A brief introduction to a widely-used lossless compression codec for audio; gave me background on probably the only serious competitor to PCM in terms of storage quality.

Keiser, Bernhard E., and Eugene Strange. *Digital Telephony and Network Integration.* New York: Springer Science+Business Media, 1985.

- One of the most readable overviews of the fundamentals of digitization and waveform coding, though date of publication means it largely addresses digital transmission (e.g. telephone, radio) rather than digitization for storage or other topics more applicable to present-day archival work.

"Knowledge Base," *High Definition Multimedia Interface,* 2017. https://www.hdmi.org/learningcenter/kb.aspx?c=11. Accessed 17 Oct. 2017.

- One of the resources that included pulse-code modulation in discussions of industry specifications; many standards are proprietary or closed, so even the mention of PCM was helpful evidence.

Kromer, Reto. "Re: [AMIA-L] Audio and Video Digitization Specifications," *AMIA-L* listserv, 27 Apr. 2016.

- A short email message, but one that served as a justification for the high end of PCM sampling rates.

Library of Congress, *Sustainability of Digital Formats* website (pages listed below)

- Excellent technical overview of pulse-code modulation (along with a wide range of related subtypes, codecs, and wrappers) in relation to the archival world, with an emphasis on stability and standards. Pages typically include a list of related topics and sources for further research.

"Advanced Audio Coding (MPEG-2)" *Sustainability of Digital Formats,* 17 Feb. 2017, https://www.loc.gov/preservation/digital/formats/fdd/fdd000036.shtml, accessed 6 Dec. 2017.

"Audio Interchange File Format (AIFF)." 27 July 2017, https://www.loc.gov/preservation/digital/formats/fdd/fdd000342.shtml. Accessed 16 Oct. 2017.

"Broadcast WAVE Audio File Format, Version 1." 27 July 2017, https://www.loc.gov/preservation/digital/formats/fdd/fdd000356.shtml. Accessed 6 Dec. 2017.

"Linear Pulse Code Modulated Audio (LPCM)." 16 Feb. 2017, https://www.loc.gov/preservation/digital/formats/fdd/fdd000005.shtml. Accessed 16 Oct.

2017.
"Material Exchange Format (MXF)." 28 Feb. 2017, https://www.loc.gov/preservation/digital/formats/fdd/fdd000013.shtml. Accessed 16 Oct. 2017.
"Matroska Multimedia Container." 28 Feb. 2017, https://www.loc.gov/preservation/digital/formats/fdd/fdd000011.shtml. Accessed 16 Oct. 2017.
"MP3 (MPEG Layer III Audio Encoding)." 27 July 2017, https://www.loc.gov/preservation/digital/formats/fdd/fdd000012.shtml. Accessed 6 Dec. 2017.
"PCM, Pulse Code Modulated Audio." 16 Feb. 2017, https://www.loc.gov/preservation/digital/formats/fdd/fdd000016.shtml. Accessed 16 Oct. 2017.
"WAVE Audio File Format." 27 July 2017, https://www.loc.gov/preservation/digital/formats/fdd/fdd000001.shtml. Accessed 16 Oct. 2017.

Emanuel Lorrain, "Re: Announcing CELLAR (Codec Encoding for LossLess Archiving and Realtime transmission)," *AMIA-L* listserv, 2 Dec. 2015.

- A discussion of the media types being explored for standardization by the PREFORMA Project (a European Union-funded effort to implement standards in electronic documents and audiovisual files). Interestingly, pairs the (losslessly compressed) FFV1 video codec with LPCM audio, rather than (losslessly compressed) FLAC as it is usually paired; this provides evidence that LPCM is overwhelmingly accepted in the A/V archiving community.

Nakajima, Heitaro. Oral history with William Aspray. Engineering and Technology History Wiki, http://ethw.org/Oral-History:Heitaro_Nakajima. Accessed 6 Dec. 2017.

- One of the few sources on pulse code modulation history with a candid, personal account of the technology's development. Added context to the laborious refinement of PCM as a recording technology.

Oliver, B.M., J.R. Pierce, and C.E. Shannon. "The Philosophy of PCM." *Proceedings of the IRE* vol. 36, no. 11 (Nov. 1948), 1324-1331.

- An early paper on PCM that outlined its benefits compared to the technology of the era, specifically FM broadcast; an invaluable resource to give context on its initial hype and adoption.

Olshausen, Bruno A. "Aliasing" (handout, PSC 129: Sensory Processes, University of California at Berkeley, Berkeley, CA, 10 Oct. 2000), http://redwood.berkeley.edu/bruno/npb261/aliasing.pdf, accessed 6 Dec. 2017.

- A brief introduction to aliasing, with a diagram, which was invaluable both in visualizing both aliasing and the broader concept of sampling at the heart of PCM.

Rice, Dave. "Re: More about digital video recording than you really want to know.," *AMIA-L* listserv, 12 Dec. 2012.

- A thoughtful push against unnecessarily uniform and singular standards in audio preservation with a strong argument for the use of FLAC.

"SIGSALY." *Audio Engineering Society,* 31 Jan. 2006, http://www.aes.org/aeshc/docs/recording.technology.history/sigsaly.html. Accessed 17 Oct. 2017.

- Explanation of one of the earliest (1943) audio communications applications of PCM.

Tomar, Rime Raj Singh, and Kapil Jain. "Lossless Image Compression Using Differential Pulse Code Modulation and its Application." *Institute of Electrical and Electronics Engineers 2015 International Conference on Computational Intelligence and Communication Networks* (Dec. 2015), 397-400.

- An example of further research and work being done with the PCM model in arenas other than audio; useful as an example of current work and prospective longevity of the encoding method.

Waggener, Bill. *Pulse Code Modulation Techniques: With Applications in Communications and Data.* New York: Van Nostrand Reinhold, 1995.

- This resource is aimed at electrical engineering students and those who work in PCM data transmission or recording systems. It provides a thorough, though highly technical, overview of PCM as used in telecommunications (as opposed to encoding for preservation and storage).

"Why Blu-ray?," *Blu-ray Disc.* Blu-ray Disc Association, 2017. https://us.blu-raydisc.com/latest_news/why-blu-ray/. Accessed 17 Oct. 2017.

- One of the resources that included pulse-code modulation in discussions of industry specifications; many standards are proprietary or closed, so even the mention of PCM was helpful evidence.